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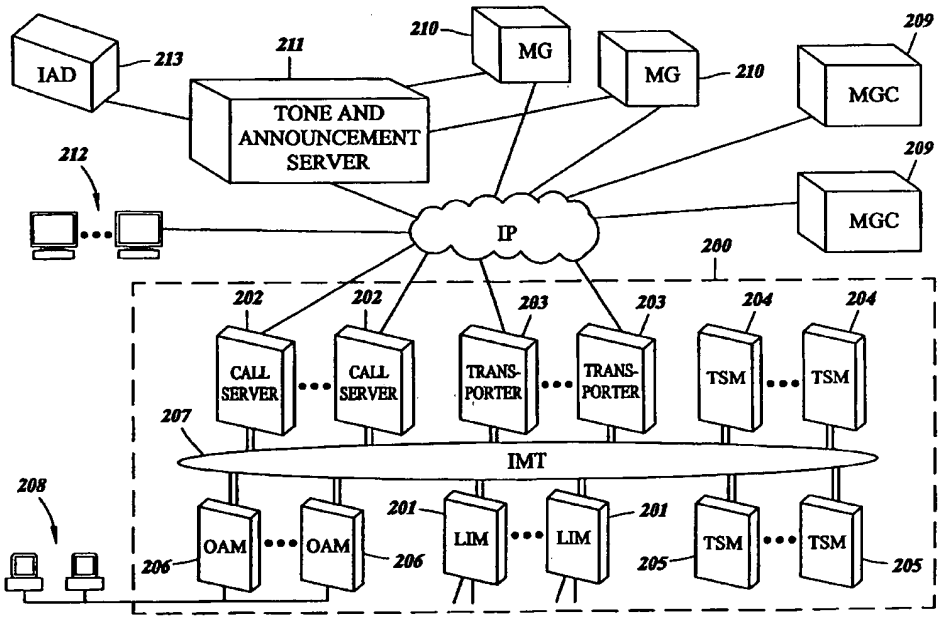

(57) Abstract: A scalable call processing node (200) includes link interface modules (201) capable of processing *n* calls per second and call server modules (202) capable of processing *m* calls per second, *n* is variable relative to *m* by changing the relative numbers of call server (202) and link interface (201) modules. In addition, call server modules (202) can perform subsecond switchover when a call server fails without requiring inter-call server transfer of call state information.

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Description

SCALABLE CALL PROCESSING NODE

Priority Application Information

This application claims the benefit of United States Patent Application Number 09/658,522 filed September 8, 2000, the disclosure of which is incorporated herein by reference in its entirety.

Technical Field

The present invention relates to methods and systems for processing call signaling messages. More particularly, the present invention relates to scalable methods and systems for processing call signaling messages.

Related Art

Voice-over-IP technology allows voice and data that was traditionally sent over time division multiplexed (TDM) connections to be sent over an Internet protocol network, such as the Internet. Voice-over-IP communication is desirable because it reduces the need for dedicated circuits between communicating entities. However, providing voice-over-IP communications requires the addition of many components to the conventional public switched telephone network (PSTN).

Figure 1 is a block diagram of a conventional solution for voice-over-IP-enabling the conventional PSTN network. In Figure 1, a calling party **100** attempts to establish voice-over-IP communication with a called party **102**. Both calling party **100** and called party **102** may utilize conventional PSTN telephones. When calling party **100** dial or keys in the telephone number for called party **102**, the dialed digits are sent to service switching point (SSP) **104**. Service switching point **104** may be a conventional PSTN end office capable of sending and receiving SS7 call signaling messages over SS7 signaling link **106** and establishing voice communications over TDM voice trunk **108**. Signal transfer point (STP) **110** routes call signaling messages to and from SSP **104** over SS7 signaling link **106**.

Continuing with the example, signal transfer point **110** routes call signaling messages to SSP **112** through SS7 signaling link **114**, STP **116**, and SS7 signaling link **118** in order to set up a call with called party **102**. SSP **112** conventionally maintains call state information for called party **102** and establishes voice communications between called party **102** and calling party **100** via the TDM voice trunk selected by SSP **104**. Thus, in the conventional case, a call can be established between calling party **100** and calling party **102** using only conventional SS7 network elements.

However, in this example, it is assumed that calling party **100** desires to establish a communication with called party **102** via IP connection **122**. In order to accomplish this IP connectivity, media gateways **124** include hardware and software for converting between TDM and IP communications. In addition, in order to set up calls using media gateways **124**, the network must also include one or more media gateway controllers. In the illustrated embodiment, the network includes six media gateway controllers **126**. Media gateway controllers **126** control media gateways **124** via IP links **128** and **130** using any number of media gateway control protocols, such as the media gateway control protocol as defined in Arango et al., RFC 2705, "Media Gateway Control Protocol (MGCP) version 1.0," (October 1999), the Megaco protocol as defined in Cuervo et al., draft-IETF-megaco-merged-01.txt, "Megaco Protocol," (May 2000), or any one of a variety of proprietary and non-proprietary protocols used for controlling media gateways.

Media gateway controllers **126** receive call signaling messages from SSPs **104** and **112** through STPs **110** and **116** and SS7 signaling links **132**. Call signaling messages received from SSPs **104** and **112** may be formatted according to the SS7 ISUP protocol. Thus, media gateway controllers **126** each include SS7 and IP communication capabilities.

Conventionally, media gateway controllers **126** have been implemented using stand-alone servers, such as the NETRA™ 1400 available from Sun Microsystems. The NETRA™ 1400 is a server that includes 1-4 Ultrasparc II processors on its motherboard, a 72.8 GB hard drive, a CD-Rom drive, and 4-6 PCI slots. A media gateway controller requires both SS7 and IP network

connections. Accordingly, two of the six possible PCI slots may hold Ethernet cards—one for communicating with media gateways and one for an administrative interface. The remaining four slots can hold SS7 cards, each of which is capable of handling two 56 kbps SS7 signaling links. Call processor functions, such as maintaining call state information, are handled by programs executing on the motherboard processors.

A problem with implementing media gateway controllers using stand-alone servers, such as Sun NETRA™ servers, is lack of a reliable way to scale the network. For example, each NETRA™ server is capable of handling at most eight 56 kbps SS7 signaling links. Adding additional SS7 signaling link capabilities requires additional NETRA™ servers. Adding additional NETRA™ servers decreases reliability of the network because of the failure rate caused by hard drives and other components of such servers. In addition, even if redundant NETRA™ servers are used to increase reliability, there is no known mechanism for performing sub-second switchover from one server to a backup server in the event that one server fails.

Another problem with using Sun NETRA™ servers to implement media gateway controller functionality is that inbound SS7 signaling link capacity is less than outbound IP signaling link capacity. For example, conventional SS7 link interface modules may be capable of processing two 56 kbps SS7 signaling links and outbound IP signaling link capacity can be 100 Mbps. This mismatch results in inefficient utilization of outbound signaling link capacity.

In light of all these difficulties associated with conventional media gateway controller solutions, there exists a long-felt need for a scalable and reliable call processing node.

Disclosure of the Invention

According to one aspect, the present invention includes a scalable call processing node having a plurality of link interface modules for receiving SS7 messages over SS7 signaling links. The link interface modules perform

call server selection based on first message parameters in the SS7 messages. The link interface modules are capable of processing at least about *n* calls per second, where *n* is an integer. The scalable call processing node also includes a plurality of call server modules. The call server modules receive SS7 messages from the link interface modules and perform call processing operations based on message parameters in the received SS7 messages. The call server modules are capable of handling at least *m* calls per second, where *m* is variable relative to *n* by changing the relative numbers of link interface and call server modules. The call processing node also includes a plurality of transporter modules operatively associated with the call server modules for formulating media gateway compatible messages based on call processing messages and forwarding the media gateway compatible messages to media gateways.

Because the call processing node according to the present invention is scalable, call processing capabilities can be increased or decreased according to network demand. In addition, outbound signaling link capacity can be more efficiently utilized by matching that capacity with inbound signaling link capacity. Finally, due to the absence of multiple mechanical components, such as disk drives, fast switchover capabilities, and decentralized power supplies, the scalable call processing node according to the present invention provides increased reliability over conventional media gateway controller solutions.

Accordingly, it is an object of the present invention to provide a call processing node that is both scalable and reliable.

Brief Description of the Drawings

A description of preferred embodiments of the present invention will now be explained with reference to the accompanying drawings of which:

Figure 1 is a block diagram of a conventional communications network in which media gateway controllers are implemented by Sun NETRA™ servers;

Figure 2 is a block diagram of a communications network including a scalable call processing node according to an embodiment of the present invention;

Figure 3 is a block diagram illustrating the scalability of a call processing node according to an embodiment of the present invention;

Figure 4 is a block diagram of exemplary call server module hardware according to an embodiment of the present invention;

Figure 5 is a flow chart illustrating exemplary steps that may be performed by call server modules in performing call server switchover according to an embodiment of the present invention;

Figure 6 is a block diagram illustrating message flow through a scalable call processing node according to an embodiment of the present invention;

Figure 7 is a block diagram illustrating exemplary call tables used by a call server module according to an embodiment of the present invention;

Figure 8 is a block diagram illustrating trunking and media gateway connections set up by a scalable call processing node according to an embodiment of the present invention;

Figure 9 is a flow chart illustrating exemplary call processing operations performed by a scalable call processing node using the call tables illustrated in Figure 7;

Figure 10 is a flow chart illustrating routing decisions made by a scalable call processing node according to an embodiment of the present invention;

Figure 11 is a block diagram of a telecommunications network including a scalable call processing node according to an embodiment of the present invention; and

Figure 12 is a block diagram of a telecommunications network including a call server module according to an alternative embodiment of the present invention.

Detailed Description of the Invention

Scalable Call Processing Node and Operating Environment

Figure 2 is a block diagram of a scalable call processing node and an exemplary operating environment for such a node according to an embodiment of the present invention. In Figure 2, scalable call processing node **200** includes a plurality of cards **201-206** connected to each other via interprocessor message transport (IMT) bus **207**. Exemplary cards that may be included in scalable call processing node **200** include link interface modules **201**, call server modules **202**, transporter modules **203**, translation service modules **204** and **205**, and operations, administration, and maintenance (OAM) modules **206**. Each of these modules will now be explained in more detail.

Link interface modules **201** may comprise SS7 link interface modules. SS7 link interface modules **201** may each include processes for sending and receiving SS7 signaling messages over SS7 signaling links and internally routing SS7 signaling messages based on one or more parameters in the SS7 signaling messages. According to the present invention, link interface modules **201** may also be capable of performing call server selection based on one or more parameters in the received SS7 signaling messages. This function will be explained in more detail below.

Exemplary link interface modules suitable for use with the present invention include two-port link interface modules, eight-port link interface modules, and twenty-four-port ATM link interface modules. Two-port link interface modules are capable of handling two 56 kbps SS7 signaling links. Eight-port LIMs are capable of handling eight 56 kbps SS7 signaling links. Finally, twenty-four-port ATM link interface modules are capable of processing 24 SS7 over ATM signaling links. The hardware associated with such link interface modules may be similar to hardware on LIMs available from Tekelec, Inc., of Calabasas, California (hereinafter, "Tekelec") in the EAGLE® STP or the IP^7 SECURE GATEWAY™ products.

Call server modules **202** include processes and databases for performing call control related functions. For example, call server modules **202** may

each include one or more databases for performing trunk selection based on parameters in a received ISUP message. Call server modules **202** may also store call state information, such as the sequence of ISUP messages received for a given call. According to an important aspect of the invention with regard to reliability, call server modules **202** preferably replicate call state information of other call servers to allow subsecond switchover in the event of failure of one of the call server modules. This function will be discussed in more detail below.

Transporter modules **203** receive messages from call server modules **202** and translate the messages between SS7 and media-gateway-controller-compatible protocols, depending on whether the destination of a message is an MG, an MGC, or an SS7 network element. For example, transporter modules **203** may each translate between ISUP and one or more of the following protocols:

MGCP, as defined in any one of the above-described IETF or RFC documents;

Session initiation protocol (SIP), as defined in Handley et al., RFC 2543, "SIP: Session Initial Protocol" (March 1999);

Transport adapter layer interface (TALI), as described in, "Transport Adapter Layer Interface 2.0 Technical Reference," Tekelec (May 2000); and

Tone and announcement server (TAS), as defined by one or more protocols for communicating with a tone and announcement server.

Translation service modules **204** may include databases and processes for performing number portability translations, such as local or mobile number portability translations. For example, TSM modules **204** may be configured to perform triggered number portability translations in response to TCAP queries received from end offices or triggerless number portability translations in response to ISUP messages received from end offices.

Functionality for performed triggered number portability translations is described in "Feature Guide LNP LSMS," PN/910-1598-01, Tekelec, Rev. A (January, 1998). Functionality for performing triggerless number portability is described in U.S. Patent Application Serial No. 09/503,541, filed February 14, 2000.

Translation service modules **205** may include databases and processes for translating from national ISUP versions to a universal ISUP protocol. For example, translation service modules **205** may receive messages formatted in Japanese ISUP, ANSI ISUP, or any other national ISUP version. Translation service modules **205** translate the national ISUP versions into a universal ISUP version understood by transporter modules **203** and IP nodes in the IP network. The universal ISUP version is referred to herein as the normalized call control protocol (NCCP). Translation service modules **205** may also include processes and databases for translating between the normalized call control protocol to a national ISUP version. For example, if a normalized call control protocol message is received by translation service modules **205**, translation service modules **205** may translate the message to the appropriate national ISUP version based on the destination of the message.

OAM modules **206** allow provisioning and maintenance of the remaining modules of scalable call processing node **200**. For example, OAM modules **206** may include serial interfaces for communication with external user terminals **208** to allow provisioning of databases in scalable call processing node **200**.

In the embodiment illustrated in Figure 2, scalable call processing node **200** communicates with a variety of other network entities. For example, in the illustrated example, scalable call processing node **200** communicates with media gateway controllers **209**, media gateways **210**, tone and announcement server **211**, peripheral interface system **212**, and integrated access device **213**. Each of these elements will now be discussed in more detail.

MGCs **209** control media gateways **210** via one of the media gateway control protocols discussed above. Transporter modules **203** may communicate with MGCs **209** via any suitable protocol, such as ISUP or SIP. Accordingly, MGCs **209** may include functionality for converting from other telephony protocols, such as ISUP or SIP, to the appropriate media gateway control protocol. MGCs **209** also store call state information for setting up and tearing down connections in media gateways **210**. An example of an external MGC suitable for use with embodiments of the present invention includes any of the Sun NETRA™-based systems described above.

Media gateways **210** perform the functions of conventional media gateways described above. For example, media gateways **210** translate between circuit-switched and packet-switched communications to allow voice and data communications over an IP network. Media gateways **210** may be controlled by media gateway controllers **209** external to scalable call processing node **200** or by call server modules **202** that are internal to scalable call processing node **200**. Exemplary media gateways suitable for use with embodiments of the present invention include the Model No. AS5300 media gateways available from Cisco Systems, Inc.

Tone and announcement server **219** plays tones to telephony users in response to predetermined network conditions. For example, tone and announcement server **219** may play normal busy tones, fast busy tones, and recorded announcements to end users. An exemplary tone and announcement server suitable for use with the present invention includes any of the TAS servers available from Radisys Corporation or Cognitronics Corporation.

Peripheral interface system **220** provides a management network for monitoring communications between the elements illustrated in Figure 2. For example, peripheral interface system **220** may allow provisioning of databases in any of the elements illustrated in Figure 2, software updates, CDR generation and analysis, billing, etc. Exemplary peripheral interface system components for CDR collection and analysis include the CDR generator as described in commonly-assigned copending U.S. Patent

Application No. 09/537,075, filed March 28, 2000, the disclosure of which is incorporated herein by reference in its entirety. Exemplary peripheral interface system components for database provisioning and software updates include a standard server, such as a Java user interface server.

Integrated access device **213** provides end user access to an IP network. For example, integrated access device **213** may allow end user telephone handsets and end user computer terminals to access the IP network. Integrated access device **213** may communicate with tone and announcement server **219** via an ATM signaling link. Integrated access device can be used as a substitute for public branch exchange (PBX) systems used in conventional telephone networks. An exemplary integrated access device suitable for use with embodiments of the present invention is the ClariNet or the Piccolo available from Woodwind Communications.

Scalability

Figure 3 is a block diagram illustrating the scalability of scalable call processing node **200** according to an embodiment of the present invention. In Figure 3, scalable call processing node **200** includes first and second shelves **301** and **302**. Each shelf is a mechanical structure in a telecommunications network equipment housing. Each shelf is capable of holding a plurality of modules or cards. As used herein, the terms "modules" or "cards" refer to printed circuit boards that are removably connectable to IMT bus **207** and that are physically housed in shelves, such as shelves **301** and **302**. Scalable call processing node **200** preferably utilizes the same internal architecture with regard to shelves and IMT bus **207** as the EAGLE® STP available from Tekelec. The EAGLE® STP is capable of holding up to 16 shelves with a maximum of 16 cards per shelf. Therefore, like the EAGLE® STP, scalable call processing node **200** is preferably scalable up to 16 shelves per system wherein each shelf is capable of holding up to 16 cards, for a total of 16^2 or 256 total cards. In the illustrated embodiment, each of the shelves **301** and **302** is capable of housing a maximum of 16 cards.

In the illustrated embodiment, first shelf **301** includes two link interface modules **201**, and second shelf **302** includes four link interface modules **201**. Thus, the system illustrated in Figure 3 includes a total of six SS7 link interface modules. For purposes of the present example, it is assumed that each link interface module comprises an eight port LIM capable of handling eight 56 kbps SS7 signaling links. Since SS7 data is transmitted serially, each byte includes eight data bits, plus a start bit and a stop bit, for a total of 10 bits. In addition, for purposes of this example, it is assumed that the ISUP messages required to set up and tear down a call require an average of 500 bytes. Accordingly, the following expression illustrates the incoming call processing capacity of scalable call processing node **200** illustrated in Figure 3:

$$\text{capacity} = (6\ \text{LIMs})\left(\frac{8\ \text{ports}}{\text{LIM}}\right)\left(\frac{56\ \text{kbits/s}}{1\ \text{port}}\right)\left(\frac{1\ \text{data byte}}{10\ \text{bits}}\right)\left(\frac{1\ \text{call}}{500\ \text{bytes}}\right)(.4) \qquad (1)$$
$$= 537.6\ \text{calls/sec}$$

In Equation 1, the number of calls processed by the LIMs illustrated in the scalable call processing node **200** illustrated in Figure 3 is discounted by a factor of .4 since SS7 signaling links are usually only operated at 40% capacity. Thus, LIMs **303-308** illustrated in Figure 3 are capable of handling 537 calls per second.

Scalable call processing node **200** illustrated in Figure 3 includes eight call server modules **202** for performing call server functions. Each call server module may be capable of handling a maximum of from about 100 to about 400 calls per second using currently available call server hardware, which will be discussed in more detail below. Thus, since scalable call processing node **200** includes eight call server modules, and each call server module is capable of processing from about 100 to 400 calls per second, the call server processing capacity of scalable call processing node **200** is from about 800 calls per second to about 3200 calls per second. A call processing capability of 3200 calls per second greatly exceeds the capacity of any media gateway controller presently known. For example, in a press release dated August 2, 2000, Sonus Networks claimed that their PSX6000™ soft switch achieved 1650 calls per second in network tests.

This is the highest number presently know and can be greatly exceeded by a scalable call processing node according to the present invention.

Finally, scalable call processing node **200** includes two transporter modules **203** for sending messages to the media gateway controllers. Since transporter modules send messages over IP signaling links and are not required to maintain call state information, the transporter modules are typically not a bottleneck to system call processing performance. For example, using currently available Ethernet-based data communication modules available from Tekelec, transporter modules **203** are each capable of sending messages at a rate of about 100 Mbps, which results in a total call processing capacity of 20,000 calls per second.

The present invention is not limited to the embodiment illustrated in Figure 3. Figure 3 simply illustrates a two-shelf system capable of handling about 537 calls per second. As stated above, using the currently available EAGLE® architecture, one call processing node can have up to 16 shelves having a maximum of 16 cards or modules per shelf. Such a system could include up to 256 cards including six OAM cards **206**. Accordingly, an alternative embodiment of the invention may include 83 eight port LIMs, for a total inbound call processing capacity of 3000 calls per second. In such an embodiment, at least eight 400 call-per-second call server modules may be included to handle the incoming calls. Finally, one transporter module may be included to provide the required outbound call translation rate. Thus, scalable call processing node **200** may be capable of processing 3000 or more calls per second, simply by adding additional call server and link interface modules. Proof of the call processing capability is illustrated by the following equations:

$$\text{inbound capacity} = (83\text{LIMs})\left(\frac{8\text{ ports}}{\text{LIM}}\right)\left(\frac{56\text{ kbits/s}}{1\text{ port}}\right)\left(\frac{1\text{ data byte}}{10\text{ bits}}\right)\left(\frac{1\text{ call}}{500\text{ bytes}}\right)(.4) \qquad (2)$$

$$= 3000\text{ calls/sec}$$

$$\text{call server capacity} = (8 \text{ call servers})\left(\frac{400 \text{ cps}}{1 \text{ call server}}\right) = 3200 \text{ calls/sec} \quad (3)$$

The call processing capability of a transporter module ranges from about 3000 to about 10,000 calls per second. In a preferred embodiment of the invention, the number of transporter modules for a given anticipated call volume is preferably doubled for load sharing and failover capabilities. Thus, if the required number of transporter modules for a given anticipated call volume is *n*, the number of transporter modules is preferably *2n*.

Thus, it is apparent from equations (2) and (3) that the call processing capabilities of scalable processing node **200** can be extended to 3000 or more calls per second.

Module Hardware

Figure 4 is a block diagram of exemplary module hardware suitable for use for LIMs **201**, call server modules **202**, and transporter modules **203** according to an embodiment of the present invention. For purposes of explanation, Figure 4 illustrates exemplary hardware for a call server module **202**. Hardware for other modules is similar to that illustrated in Figure 4. In Figure 4, call server module **202** includes application processor **400**, communication processor **401**, and interprocessor memory **402**.

Application processor **400** executes programs for performing call processing operations, such as storing call state information, formulating call processing messages in response to other call processing messages received from communication processor **401**. Communication processor **401** sends and receives messages via IMT bus **207**. Interprocessor memory **402** is shared by application processor **400** and communication processor **401**. Because processors **400** and **401** utilize shared memory, the efficiency of call processing module **202** is increased. An exemplary commercially available microprocessor suitable for use as application processor **400** is the K6-2 available from AMD Corporation. An exemplary microprocessor

suitable for use as communication processor **401** is the K6-2 available from AMD Corporation.

In the illustrated embodiment, call server module **202** preferably includes its own power supply **403**. Such a power supply may be configured to provide power to processors **400** and **401** and memory **402**, as well as other circuitry of call server module **202**. Power supply **403** preferably received its power from a system power supply, which is preferably an uninterruptible power supply (UPS). Any suitable commercially available power supply for providing power at logic levels can be used for power supply **403**. What is important for purposes of the present invention is that each call server module preferably includes its own power supply. Thus, if one power supply fails, only one call server module will fail. This is in contrast to the conventional solution where media gateway controllers are implemented by Sun NETRA™ servers. In those systems, if the power supply fails, all of the media gateway controller functionality of that system fails.

Subsecond Switchover

According to another aspect, the present invention includes methods and systems for performing subsecond switchover of call servers in the event that one call server fails. Figure 5 illustrates exemplary steps that may be performed by a call processing node according to an embodiment of the present invention in performing subsecond switchover. Referring to Figure 5, in step **ST1**, scalable call processing node **200** establishes one call server module as a primary call server module and another call server module as a backup call server module. The decision as to whether a call server will be a primary or a backup module may depend on any suitable criteria, such as the memory address at which the call server software is located. For example, the call server software having the lowest memory address may be designated the primary call server module.

In step **ST2**, one of the LIMs **201** illustrated in Figure 1 receives call signaling messages for a call and sends the call signaling messages to the primary and backup call server modules. For example, each LIM may make

a copy of the original message and send the original along with the copy to two different call server modules. In an alternative embodiment, each LIM may send the original message to the primary call server and the primary call server may send a copy to the backup call server. In step **ST3**, the primary and backup call server modules each store state information for the call. In this context, state information includes any information required to set up, maintain or tear down a call, such as messages received, trunk information, linkset information, media gateway endpoint information, etc. Exemplary call state information that may be replicated in primary and backup call server modules will be described in more detail below with regard to Figure 7.

Although both call servers store the call state information, only the primary call server module actually sends call signaling messages related to the call to outbound communications modules. In step **ST4**, the backup call server module determines whether the primary call server module has failed. Referring back to Figure 4, this determination may be made by communication processor **401** associated with the secondary call server. For example, communication processor **401** of the backup call server may monitor a heartbeat message from the communication processor of the primary call server. If the heartbeat message fails to arrive within a predetermined time period, communication processor **401** of backup call server module may notify application processor **400** of the backup call server of the failure of the primary call server module. In step **ST5**, application processor **400** of the backup call server module switches to perform primary call server functions, such as sending the appropriate call signaling messages to a media gateway to set up a connection in the media gateway.

Thus, as illustrated in Figure 5, switchover may occur between call sever modules connected to the IMT bus. Because the primary and backup call server modules each receive copies of all of the call signaling messages associated with a call, because both call servers retain call state information for the call, and because the call servers are connected to a high-speed IMT bus, subsecond switchover of call servers can be achieved. Unlike the prior

art where it is necessary to transfer call state information from one media gateway controller to another media gateway controller when one media gateway controller fails, this transfer is not necessary in the present embodiment. Communications can resume without transfer of call state information due to the redundant storage thereof by call server modules according to an embodiment of the present invention.

Scalable Call Processing Node Internal Architecture and Message Flow

Figure 6 is block diagram illustrating internal architecture and message flow for a scalable call processing node according to an embodiment of the present invention. For purposes of illustration, it is assumed that the incoming message is an ISUP message and the outgoing message is a media-gateway-compatible message.

In Figure 6, scalable call processing node **200** includes LIM **201**, call server module **202**, transporter module **203**, translator module **205**, and IMT bus **207**. It is understood that although scalable call processing node **200** includes a single LIM, call server, translator, and transporter module, any number of these modules may be included within the scalable call processing node **200**. One module of each type is shown to simplify the explanation of the message flow.

LIM **201** includes SS7 layer 1 and 2 process **600** for performing SS7 layer 1 and 2 functions on incoming messages. I/O queue **601** stores messages for processing by higher SS7 layer processes. Message handling and discrimination (HMDC) process **602** performs discrimination of incoming messages to determine whether the messages are addressed to scalable call processing node **200** or whether the messages should be through-switched. Such a determination may be made based on a destination point code value in the incoming SS7 messages. Message handling and routing (HMRT) process **603** internally routes messages that are directed to scalable call processing node **200**. According to the present invention, HMRT process **603** may be provisioned to perform call server selection based on one or more parameters in the SS7 call signaling messages. Exemplary

parameters that may be used to perform call server selection are the OPC, DPC, and CIC codes in an incoming SS7 message.

Call server module **202** includes call processor **604** and one or more call tables **604A** for maintaining call state information and setting up a connection using media gateway. Figure 7 illustrates exemplary call tables **604A** that may be stored in memory on call server module **202**. Referring to Figure 7, call tables **604A** include a translation table **700**, a routing table **701**, a signaling table **702**, an endpoint table **703**, a connection table **704**, and a state table **705**. Each of these tables may be variously configured. In the illustrated embodiment, translation table **700** maps dialed digits to trunk groups. Routing table **701** maps trunk groups to media gateways and SS7 routing sets. Signaling table **702** maps SS7 routing sets to destination point codes and linksets. Routing table **701** and signaling table **702** are used to generate SS7 call signaling messages relating to a call. Endpoint table **703** and connection table **704** contain information for establishing a connection in a media gateway. Finally, state table **705** stores call state information for each endpoint in a media gateway. The use of tables **700-705** to set up a call will now be described in more detail.

Figure 8 illustrates exemplary trunking and connections in a voice-over-IP network including a scalable call processing node according to an embodiment of the present invention. In Figure 8, end office **800** is connected to end offices **801-803** by media gateway **804**. More particularly, trunk groups 4 and 5 connect end office **800** to media gateway **804** and trunk groups TG1-TG3 connect media gateway **804** to end offices **801-803**. Each trunk group includes a plurality of channels, which are identified by CIC codes unique to each end office. STPs **805** and **806** route call signaling messages between end office **800** and end office **801**. Finally, scalable call processing node **200** sets up, maintains, and tears down connections in media gateway **804**.

Figure 9 illustrates exemplary steps that may be performed in setting up a call between an end user connected to end office **800** and another end user connected to end office **801** illustrated in Figure 8 using call tables

604A illustrated in Figure 7. Referring to Figure 9, in step ST1, scalable call processing node 200 receives an ISUP IAM message from end office 800. The parameters in the ISUP IAM message may be as follows:

OPC=1-1-7, DPC=2-1-1, CIC=3, CIgPty=919-460-5500, CIdPty=919-787-8009.

In step ST2, scalable call processing node 200 determines the incoming port on media gateway 804 using the OPC, DPC, and CIC codes in the message. In this example, it is assumed that the incoming port number corresponding to the OPC, DPC, CIC combination is 1002. In step ST3, call processing node 200 determines a trunk group for the outgoing trunk using the called party number and translation table 700 in Figure 7. In Figure 7, translation table 700 indicates that the called party digits 919-787-xxxx corresponds to trunk group TG1.

In step ST4, scalable call processing node 200 selects an outgoing trunk in trunk group 1. This selection may be performed by choosing the next available circuit within the trunk group. In this example, it is assumed that the trunk corresponding to CIC code 2 is the first available trunk in the trunk group. In step ST5, scalable call processing node 200 formulates an MGCP CreateConnection message and sends the message to the media gateway. This message may be formulated by transporter module 703 illustrated in Figure 6 based on parameters received from call server module 202. In order to determine the parameters that must be included in the CreateConnection message, call server module 202 may access endpoint table 703 illustrated in Figure 7. In this example, since the trunk group is TG1, the OPC is 1-1-10, and the CIC code is 2, the outgoing port on media gateway 804 is port number 2533. The connection ID assigned to the connection in media gateway 804 is 0. Accordingly, scalable call processing node 200 formulates an MGCP CreateConnection message with the following parameters:

ID=0, EP_ID=1002, SEC EP_ID=2533.

In response to the MGCP CreateConnection message, media gateway **804** returns connection identifiers corresponding to each end of the connection in media gateway **804**. In this example, the connection identifier for the first endpoint is assumed to be 89 and the connection identifier corresponding to the second endpoint of the connection is 90. These parameters are stored in connection table **704** illustrated in Figure 7.

In step **ST6**, scalable call processing node **200** determines data to be included in an IAM message sent out to end office **801** to select the outgoing trunk between end office **801** and media gateway **804**. In order to make this determination, scalable call processing node **200** uses routing table **701** and signaling table **702** illustrated in Figure 7. Referring to routing table **701**, if the trunk group is TG1, the SS7 routing set is RS1. Referring to signaling table **702**, if the routing set is RS1, the destination point code is 1-1-10, and the linksets are LS1 and LS2. In step **ST7**, scalable call processing node **200** sends the IAM message to end office **801**. In this example, the parameters that may be included in the IAM message are:

OPC=2-1-1, DPC=1-1-10, CIC=2, CIgPty=919-460-5500,
CIdPty=919-787-8009.

The IAM message instructs end office **801** to set up a trunk corresponding to CIC code 2.

In step **ST8**, scalable call processing node **200** updates call state information in state table **705**. State table **705** preferably contains an entry for each endpoint. In the illustrated example, the endpoint corresponding to port 1001 in media gateway **804** is in the state received IAM, indicating that an IAM message has been received for that endpoint. Endpoint ID **2533** is in the state generated IAM and waiting for ACM. Step **ST8** is preferably performed any time a message relating to a connection is sent or received. The state information stored in table **705** is not to be confused with the state

information exchanged between primary and backup media gateways described above with respect to Figure 7, which may include any or all of the information contained in call tables **604A**.

In step **ST9**, scalable call processing node **200** receives an address complete message from end office 1-1-10. In step **ST10**, scalable call processing node **200** forwards the address complete message (ACM) to end office **800**. When the called party answers the call, an answer (ANM) message is sent from end office **801** through scalable call processing node **200** to end office **800**. The ANM message follows the same path as the ACM message. Once the ANM message is received, a voice connection is established between end office **800** and end office **801** through media gateway **804**. Thus, Figures 7-9 illustrate the use of call tables **604A** in setting up a call using a media gateway.

Referring back to Figure 6, transporter module **203** includes upper layer protocol converter **605** for converting between SS7 and a media-gateway-compatible or media-gateway-controller-compatible protocol, such as MGCP, SIP, or any of the other protocols discussed above. Transporter module **203** also includes SS7-to-IP converter **606** for converting between SS7 and IP address schemes. Finally, translator module **205** includes ISUP translator **607** for converting from national to normalized ISUP and vice versa.

The internal operation of scalable call processing node **200** illustrated in Figure 6 will now be explained with reference to the flow chart illustrated in Figure 9. In Figure 9, in step **ST1**, LIM **201** receives an ISUP message. Such a message may be an initial address message (IAM), an address complete message (ACM), an answer message (ANM), a release message (REL), or a release complete message (RLC). In this example, it is assumed that an IAM message is received. In step **ST2**, LIM **201** illustrated in Figure 6 determines whether the message should be through-switched. As stated above, this determination may be made based on the destination point code in the message. In step **ST3A**, if the message is to be through-switched, HMDC process **602** in LIM **201** routes the message to the appropriate

module for outbound processing. In this example, it is assumed that the message is not a message that is to be through-switched.

In step **ST4**, HMRT process **603** in LIM **201** performs call server selection based on the OPC, DPC, and CIC parameters in the received SS7 message. In step **ST5**, HMRT process **603** routes the message to the appropriate call server. In step **ST6**, call processor **604** performs call processing operations in response to the received SS7 message. Exemplary call processing operations that may be performed include the operations relating to setting up a connection in media gateway **804** described with respect to Figures 7-9.

An additional function that may be performed by call processor **604** is determining whether translation is required. As used herein, translation refers to translation to or from a normalized ISUP protocol. In order to make this determination, call processor **604** may determine the ISUP message party end office based on one or more parameters, such as DPC in the received ISUP message. In step **ST8**, if translation is required, call processor **604** may forward the message to ISUP translator **607**, where a translation is performed, and receive a translated message from translator **607**.

In step **ST9**, call processor **604** routes either the translated or the non-translated call signaling message to transporter module **203** for outbound processing. In step **ST10**, upper layer transport module **605** determines the protocol of the destination media gateway and translates the upper layer portion of the received message to the upper layer protocol of the destination. For example, upper layer transport module may translate the message from ANSI ISUP to MGCP. Lower layer transport processor **606** converts the lower level portion of the message to Internet protocol. Transporter module **203** then routes the message to an appropriate media gateway. Thus, Figure 7 illustrates internal routing decisions performed by scalable call processing node **200**.

Call Setup Using Media Gateway Controllers and Scalable Call Processing Node

Figure 11 is a network diagram illustrating call setup using scalable call processing node **200** and a media gateway controller **210** according to an embodiment of the present invention. In the example, steps for setting up a call between an end user associated with SSP **800** and an end user associated with SSP **802** will be described. The call is set up between media gateways **804** and **806**. Call signaling messages for the call are routed through signal transfer points **808** and **810**. The circled numerals in Figure 11 refer to steps required for call setup which will now be described.

In step **ST1**, SSP **800** receives dialed digits from a calling party. In this example, it is assumed that the calling party number is 919-460-5500 and the called party is 219-884-8009. SSP **800** selects a trunk for voice communications by specifying circuit identification code of 50. SSP **800** then formulates and sends an IAM message to SSP **802** controlling the other end of the trunk. The OPC in such a message is 1-1-7, the DPC is 2-2-1, and the CIC is 50. In step **ST2**, the IAM message is sent to STP **808** for SS7 routing. STP **808** routes the IAM message to scalable call processing node **200**. An HMRT process on the receiving LIM of scalable call processing node **200** selects a call server module and forwards the message to the selected call server module.

In step **ST4**, scalable call processing node **200** sends an MGCP CreateConnection request to MG **804** to set up an internal connection between incoming trunk from SSP **800** and the outgoing connection to media gateway **806**. In this example, the outgoing connection to media gateway **806** may be IP, ATM, frame relay, TDM, or any other packet-based protocol for carrying the media stream between the called and calling parties. Media gateway **804** uses the information in the CreateConnection message to set up an internal connection between the TDM trunk connected to SSP **800** and the IP "trunk" connected to MG **806**. In step **ST5**, media gateway **804** sends a response to scalable call processing node **200** indicating that the CreateConnection operation was successfully performed.

In step **ST6**, scalable call processing node **200** formulates a new IAM message directed to MGC **240** having the point code 1-1-8. In step **ST7**, STP **808** forwards the new IAM message through the network. In step **ST8**, MGC **210** receives the IAM message from its SS7 stack. In step **ST9**, MGC **210** generates a CreateConnection message requesting MG **806** to set up an internal connection between two trunks, the incoming trunk from media gateway **804** and the outgoing trunk to SSP **802**. In response to the CreateConnection message, media gateway **806** performs the steps necessary to set up the internal connection between the IP trunk connected to MG **804** and the TDM link to SSP **802**. In step **ST10**, MG **806** acknowledges to the CreateConnection message.

In response to the CreateConnection acknowledgement message, in step **ST11**, MGC **210** formulates a new IAM message and sends the new IAM message to SSP **802** having the point code 55-2-2 so that SSP **802** will set up the trunk. In step **ST12**, STP **810** forwards the IAM message to SSP **802**. In step **ST13**, SSP **802** completes the trunk setup operation.

At this point in the call, SSP **802** sends an ACM message to SSP **800**. In response to the ACM message, SSP **800** applies a ring-back message to the calling party and SSP **802** applies a ringing signal to the called party. When the called party answers the call, an ANM message is forwarded by SSP **802** to SSP **800**. Thus, Figure 8 illustrates call setup using a scalable call processing node according to an embodiment of the present invention.

Call Setup Using SIP

Figure 12 is a network diagram with identical entities to the network illustrated in Figure 11. However, in Figure 12, scalable call processing node **200** and MGC **210** exchange trunk setup messages using SIP rather than sending ISUP SS7 messages to each other through STPs **808** and **810**. The steps in Figure 12 other than steps **ST6**, **ST7**, and **ST8** are identical to those illustrated in Figure 11. Hence, a description thereof will not be repeated herein. Referring to step **ST6** in Figure 12, scalable call processing node **200** formulates a SIP message and sends the SIP message to MGC **210**. The SIP message may be an INVITE message. The SIP INVITE

message includes the outgoing trunk. Steps **ST7** and **ST8** indicate additional SIP messages that may be exchanged between scalable call processing node **200** and MGC **210** in order to set up a call between the parties. An example of a SIP INVITE message that may be formulated by scalable call processing node **200** according to the present embodiment is as follows:

INVITE sip: 19197878009@southbell.com SIP/2.0
From: sip: 19194605500@office.tekelec.com
To: sip: 19197878009@southbell.com
Call-ID: SOUTH94738299197878009@southbell.com

In response to the SIP message, MGC generates the CreateConnection message requesting MG **806** to set up a trunk connecting point code 2-1-1 and point code 55-2-2. Thus, the embodiment in Figure 12 illustrates call setup using SIP according to an embodiment of the present invention.

It will be understood that various details of the invention may be changed without departing from the scope of the invention. Furthermore, the foregoing description is for the purpose of illustration only, and not for the purpose of limitation—the invention being defined by the claims.

CLAIMS

What is claimed is:

1. A scalable call processing node for receiving SS7 messages and for formulating media-gateway-compatible messages, the scalable node comprising:

   (a) a plurality of link interface modules (LIMs) for receiving SS7 messages over SS7 signaling links and performing call server selection based on first message parameters in the SS7 messages, the link interface modules being capable of processing SS7 messages for at least about *n* calls per second, *n* being an integer;

   (b) a plurality of call server modules operatively associated with the link interface modules for receiving the SS7 messages from the link interface modules based on the call server selection and for performing call setup operations for setting up connections in media gateways, the call server modules being capable of setting up at least about *m* calls per second, *m* being an integer, wherein *n* is variable relative to *m* by changing the relative numbers of link interface and call server modules; and

   (c) a plurality of transporter modules operatively associated with the call server modules for formulating media-gateway-compatible messages based on the call processing operations, and for forwarding the media-gateway-compatible messages to media gateways.

2. The scalable call processing node of claim 1 wherein *n* is substantially equal to m.

3. The scalable call processing node of claim 1 wherein *m* is at least about 1000 calls per second.

4. The scalable call processing node of claim 1 wherein *m* is at least about 2000 calls per second.

5. The scalable call processing node of claim 1 wherein *m* is at least about 3000 calls per second.

6. The scalable call processing node of claim 1 wherein *m* is at least about 4000 calls per second.

7. The scalable call processing node of claim 1 wherein each of the LIMs is capable of handling eight 56 kbps SS7 signaling links.

8. The scalable call processing node of claim 1 comprising an interprocessor message transport (IMT) bus for interconnecting the link interface modules, the call server modules, and the transport modules.

9. The scalable call processing node of claim 8 wherein the link interface modules are adapted to send the SS7 messages to the call server modules over the IMT bus.

10. The scalable processing node of claim 8 wherein the call server modules are adapted to send the call processing messages to the transporter modules over the IMT bus.

11. The scalable call processing node of claim 1 wherein each of the call server modules comprises:

(a) a translation table for translating called party address digits to trunk groups;

(b) a routing table for assigning the trunk groups to corresponding media gateways;

(c) an endpoint table for assigning trunks in the trunk groups to port numbers on the media gateways;

(d) a connection table for storing connection information for each connection in each of the media gateways; and

(e) a state table for storing call state information for each endpoint of each connection.

12. The scalable call processing node of claim 1 wherein the SS7 messages received by the LIMs include ISUP messages.

13. The scalable call processing node of claim 1 wherein the call processing operations performed by the call server modules include selecting endpoints in a media gateway to set up a connection.

14. The scalable call processing node of claim 13 wherein the call processing operations performed by each call server include maintaining call state information for the endpoints.

15. The scalable call processing node of claim 1 wherein the media-gateway-compatible messages generated by the transport modules include CreateConnection messages.

16. The scalable call processing node of claim 8 wherein the LIMS, the call server modules, and the transporter modules comprise printed circuit boards, each of the printed circuit boards being removably coupleable to the IMT bus to increase or decrease call signaling processing capacity of the scalable call processing node.

17. The scalable call processing node of claim 16 wherein each of the printed circuit boards includes at least one microprocessor and associated memory, wherein the microprocessor associated with each of the LIMs is programmed to receive and route the SS7 messages, the microprocessor associated with each of the call server modules is programmed to perform the call processing operations, and wherein the microprocessor associated with each of the transporter modules is programmed to formulate the media-gateway-compatible messages.

18. The scalable call processing node of claim 17 wherein none of the printed circuit boards includes a disk drive.

19. The scalable call processing node of claim 1 wherein the plurality of call processing modules include:

(a) a first call server module functioning as the primary call server module for a call; and

(b) a second call server module functioning as a backup call server module for the call, wherein both the primary and backup call server modules store call state information for the call and wherein the second call server module switches to become the primary call server module in response to failure of the first call server module.

20. The scalable call processing node of claim 19 wherein the switching from backup to primary call server module occurs in less than one second.

21. The scalable call processing node of claim 19 wherein the switching occurs without transfer of call state information from the first call server module to the second call server module.

22. A scalable call processing node comprising:

(a) a link interface module for receiving SS7 call signaling messages and for performing call server selection based on at least one parameter in the SS7 messages;

(b) a first call server module for receiving the SS7 messages from the LIM and for functioning as a primary call server for a call and adapted to store state information for the call; and

(c) a second call server module for storing the state information and functioning as a backup call server for the call, wherein the second call server module is adapted to switch operation to become the primary call server for the call in response to failure of the first call server module.

23. The scalable call processing node of claim 22 wherein the switching from backup to primary call server module occurs in less than one second.

24. The scalable call processing node of claim 22 wherein the switching occurs without transfer of the call state information

from the first call server module to the second call server module.

25. The scalable call processing node of claim 22 wherein the state information includes at least one call table for storing call-related information.

26. The scalable call processing node of claim 25 wherein the at least one call table includes an endpoint table for storing endpoint information for a media gateway.

27. The scalable call processing node of claim 26 wherein the at least one call table includes a connection table for storing connection information for connections in the media gateway.

28. The scalable call processing node of claim 26 wherein the at least one call table includes a state table for storing call signaling state information for endpoints in the media gateway.

29. A method for performing call server module switchover in a scalable call processing node in response to call server module failure, the method comprising:

(a) storing call state information for a first call on first and second call server modules connected to each other via an interprocessor message transport bus;

(b) operating the first call server module in a primary call server mode and operating the second call server module in a backup call server mode;

(c) detecting failure of the first call server module; and

(d) in response to failure of the first call server module, switching the second call server module to the primary call server mode without transferring the call state information from the first call server module to the second call server module.

30. The method of claim 29 wherein storing call state information includes storing parameters extracted from a sequence of ISUP messages required to set up or tear down the first call.

31. The method of claim 29 wherein the primary call server mode includes formulating instructions for setting up or tearing down the first call and forwarding the instructions to a transporter module for translation and transport to a media gateway.

32. The method of claim 29 wherein the backup mode includes storing call state information without forwarding call processing messages to intended destinations.

33. The method of claim 29 wherein switching operation of the second call server module to the primary mode includes switching the operation within a fraction of one second.

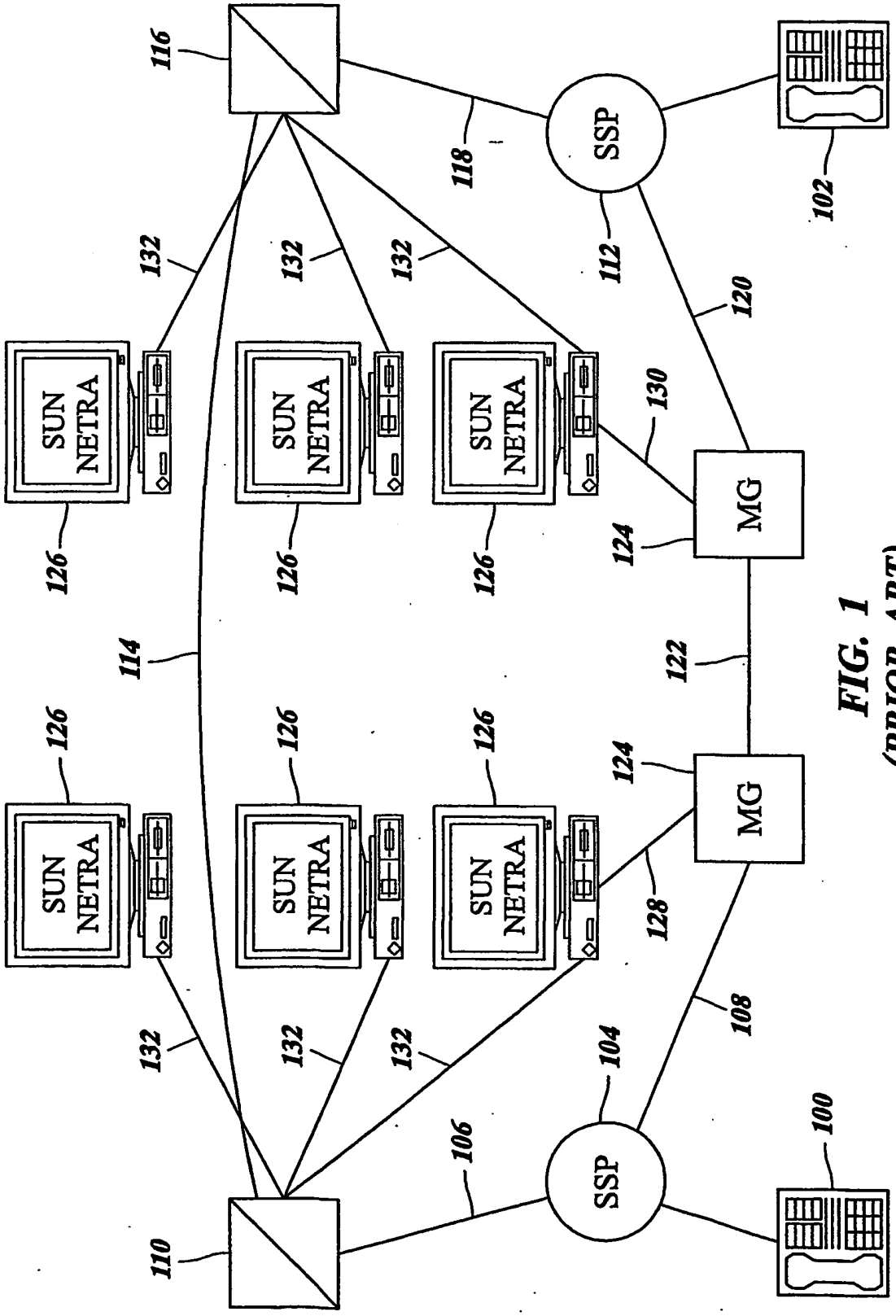


FIG. 1
(PRIOR ART)

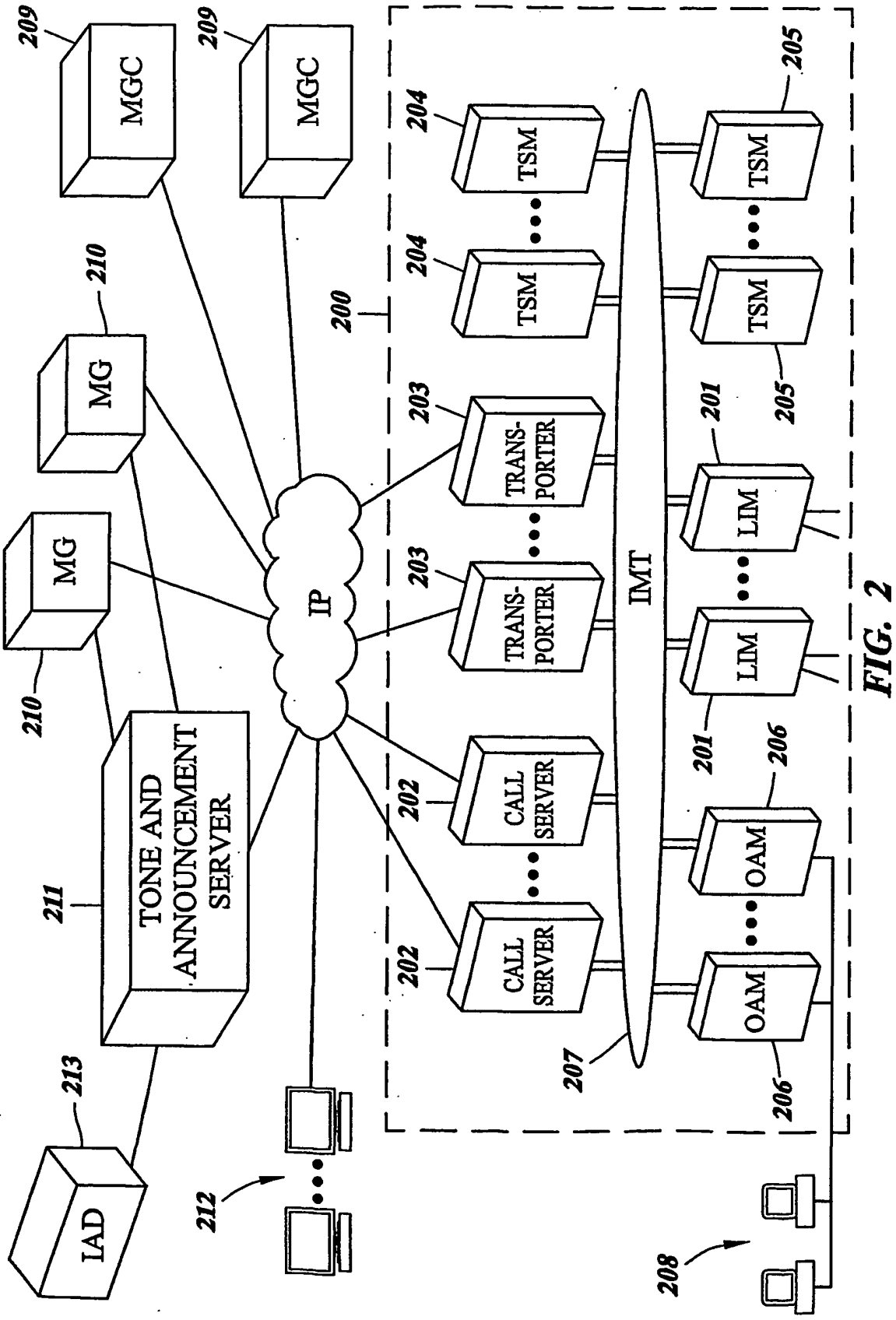

209
MGC
209
MGC
210
MG
MG
210
IP
200
204
TSM
204
TSM
203
TRANS-PORTER
203
TRANS-PORTER
202
CALL SERVER
202
CALL SERVER
207
IMT
205
TSM
205
TSM
201
LIM
201
LIM
206
OAM
206
OAM
211
TONE AND ANNOUNCEMENT SERVER
213
IAD
212
208

FIG. 2

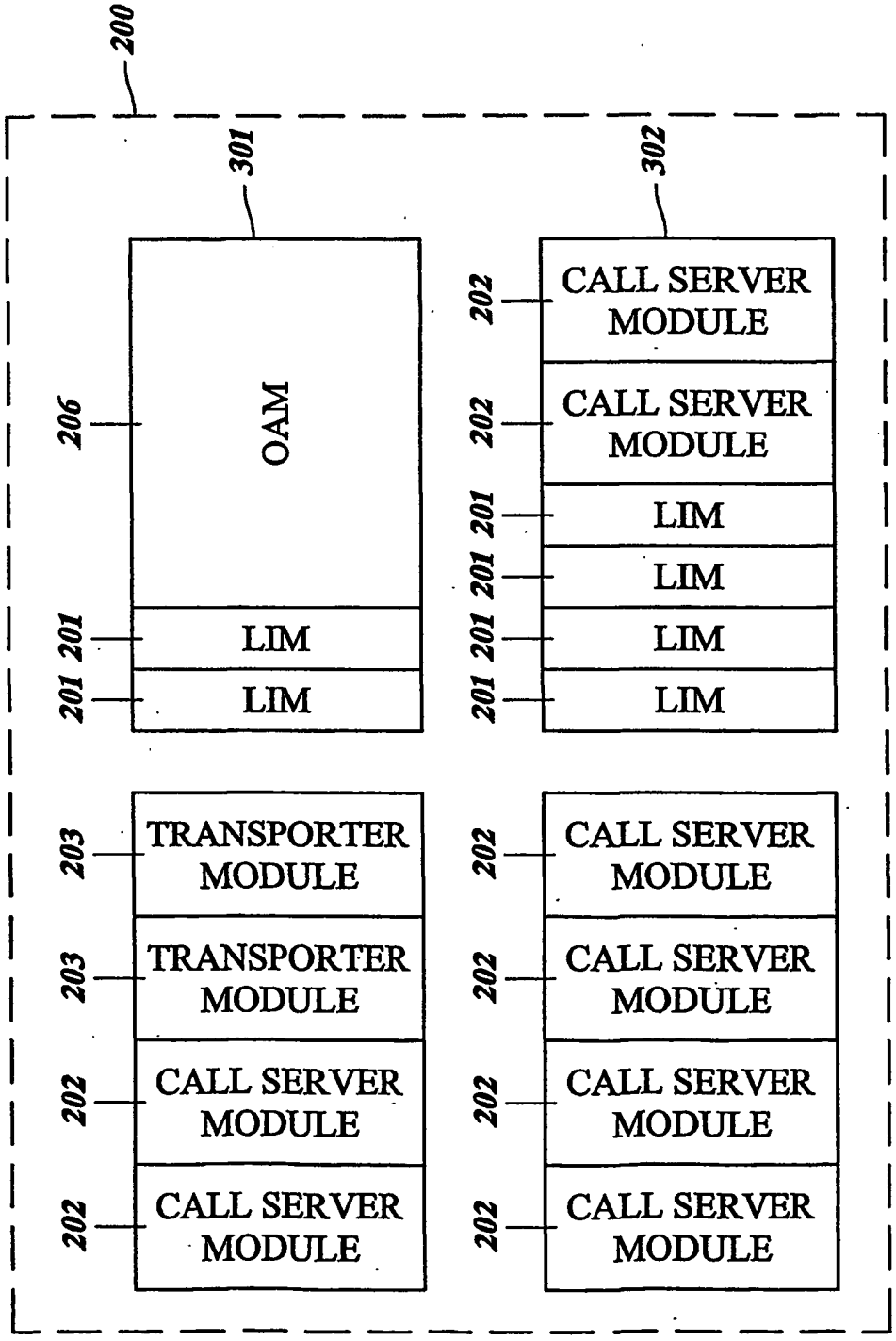

200
301
302
206
OAM
201
LIM
201
LIM
202
CALL SERVER MODULE
202
CALL SERVER MODULE
201
LIM
201
LIM
201
LIM
201
LIM
203
TRANSPORTER MODULE
203
TRANSPORTER MODULE
202
CALL SERVER MODULE
202
CALL SERVER MODULE
202
CALL SERVER MODULE
202
CALL SERVER MODULE
202
CALL SERVER MODULE
202
CALL SERVER MODULE

FIG. 3

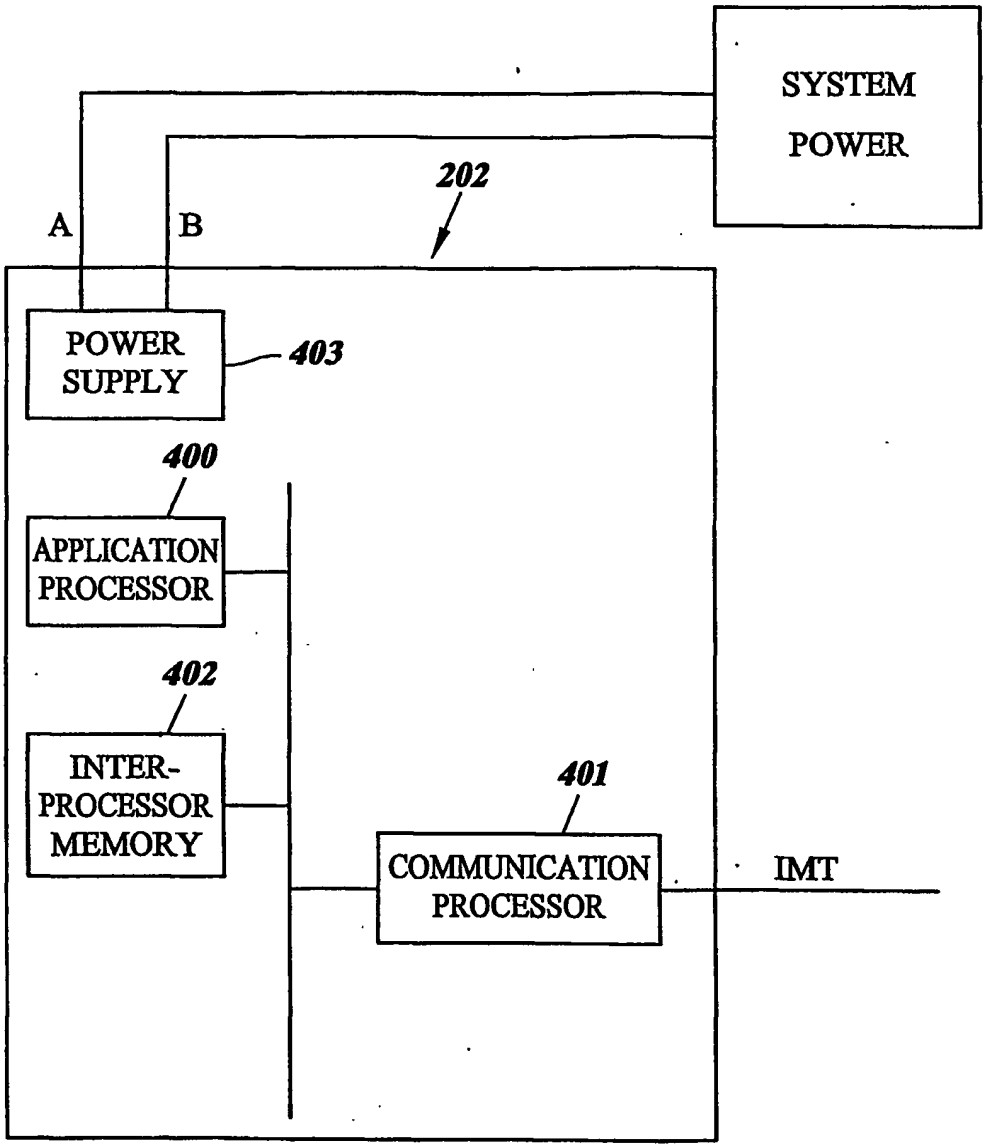

SYSTEM POWER
A
B
202
POWER SUPPLY
403
400
APPLICATION PROCESSOR
402
INTER-PROCESSOR MEMORY
401
COMMUNICATION PROCESSOR
IMT

FIG. 4

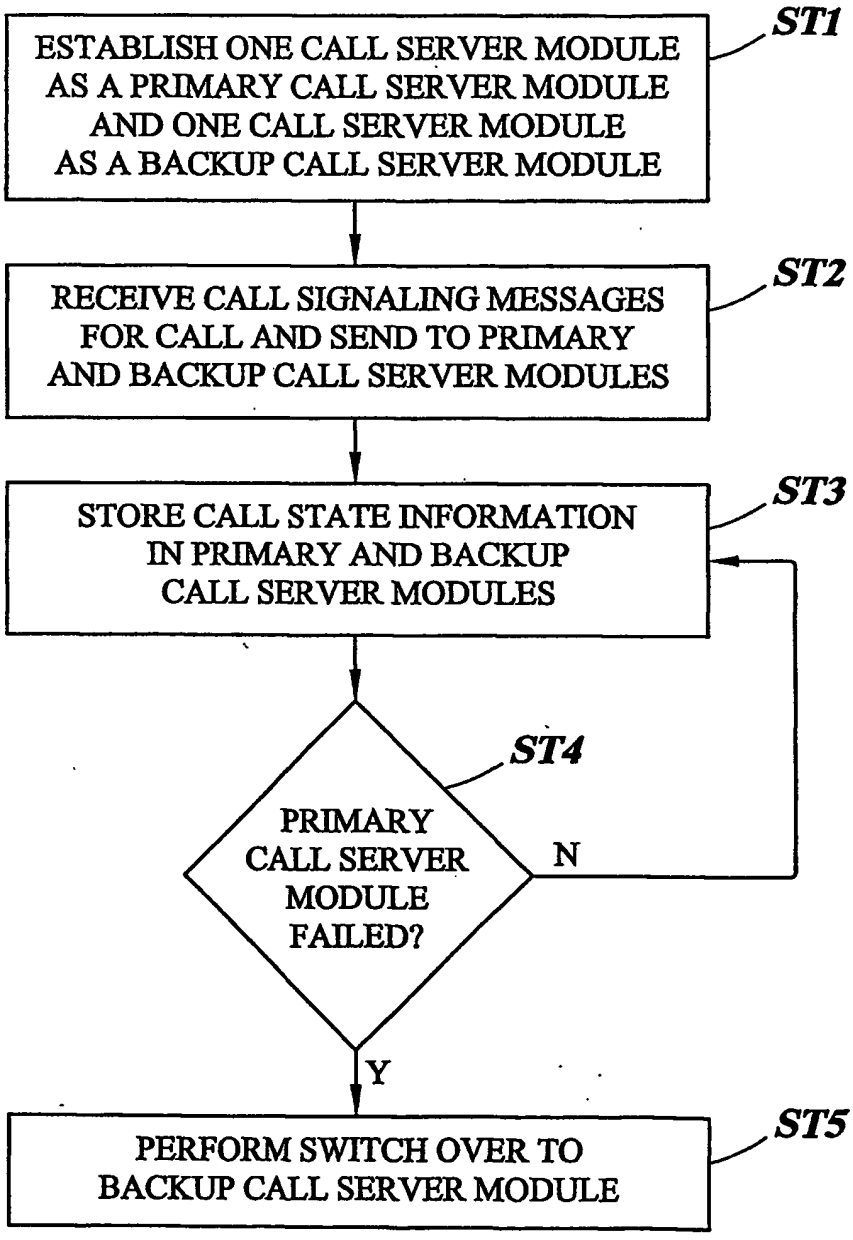

ESTABLISH ONE CALL SERVER MODULE AS A PRIMARY CALL SERVER MODULE AND ONE CALL SERVER MODULE AS A BACKUP CALL SERVER MODULE
ST1
RECEIVE CALL SIGNALING MESSAGES FOR CALL AND SEND TO PRIMARY AND BACKUP CALL SERVER MODULES
ST2
STORE CALL STATE INFORMATION IN PRIMARY AND BACKUP CALL SERVER MODULES
ST3
PRIMARY CALL SERVER MODULE FAILED?
ST4
N
Y
PERFORM SWITCH OVER TO BACKUP CALL SERVER MODULE
ST5

FIG. 5

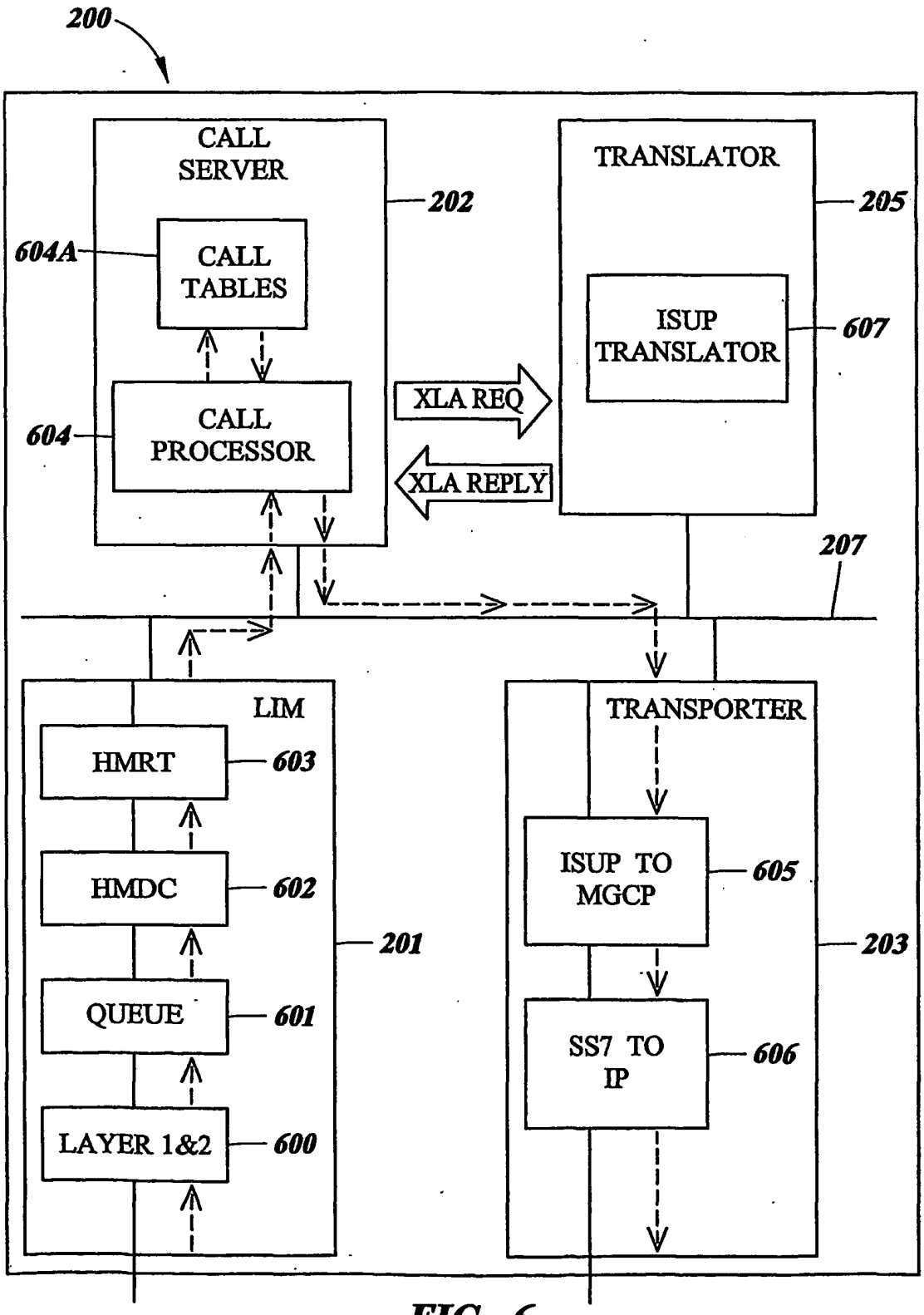
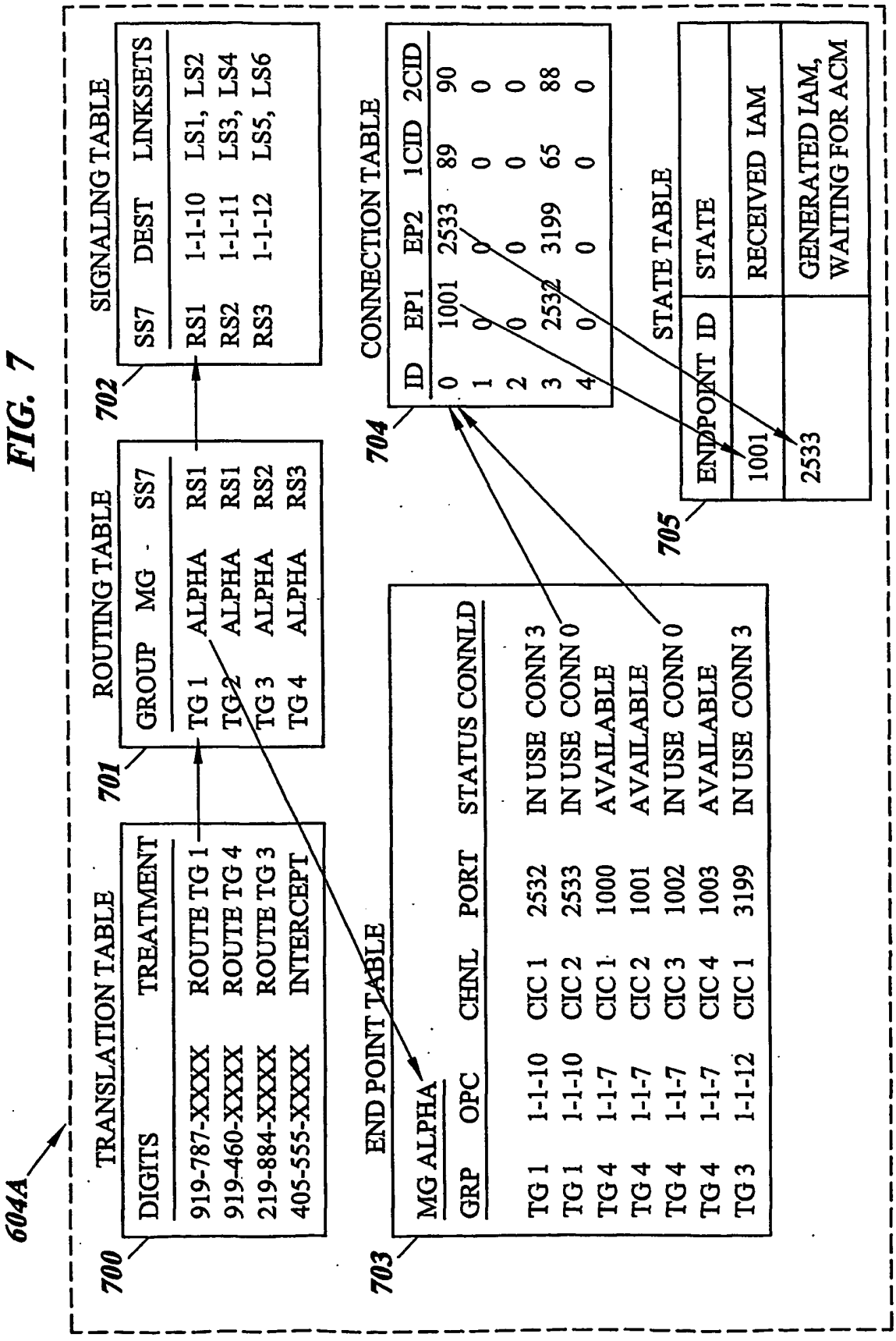

200
CALL SERVER
202
604A
CALL TABLES
604
CALL PROCESSOR
XLA REQ
XLA REPLY
TRANSLATOR
205
ISUP TRANSLATOR
607
207
LIM
HMRT
603
HMDC
602
201
QUEUE
601
LAYER 1&2
600
TRANSPORTER
ISUP TO MGCP
605
203
SS7 TO IP
606

FIG. 6

FIG. 7


604A
TRANSLATION TABLE
700
DIGITS
TREATMENT
919-787-XXXX
ROUTE TG 1
919-460-XXXX
ROUTE TG 4
219-884-XXXX
ROUTE TG 3
405-555-XXXX
INTERCEPT
ROUTING TABLE
701
GROUP
MG
SS7
TG 1
ALPHA
RS1
TG 2
ALPHA
RS1
TG 3
ALPHA
RS2
TG 4
ALPHA
RS3
SIGNALING TABLE
702
SS7
DEST
LINKSETS
RS1
1-1-10
LS1, LS2
RS2
1-1-11
LS3, LS4
RS3
1-1-12
LS5, LS6
END POINT TABLE
703
MG ALPHA
GRP
OPC
CHNL
PORT
STATUS
CONNLD
TG 1
1-1-10
CIC 1
2532
IN USE
CONN 3
TG 1
1-1-10
CIC 2
2533
IN USE
CONN 0
TG 4
1-1-7
CIC 1
1000
AVAILABLE
TG 4
1-1-7
CIC 2
1001
AVAILABLE
TG 4
1-1-7
CIC 3
1002
IN USE
CONN 0
TG 4
1-1-7
CIC 4
1003
AVAILABLE
TG 3
1-1-12
CIC 1
3199
IN USE
CONN 3
CONNECTION TABLE
704
ID
EP1
EP2
1CID
2CID
0
1001
2533
89
90
1
0
0
0
0
2
0
0
0
0
3
2532
3199
65
88
4
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0
0
STATE TABLE
705
ENDPOINT ID
STATE
1001
RECEIVED IAM
2533
GENERATED IAM, WAITING FOR ACM

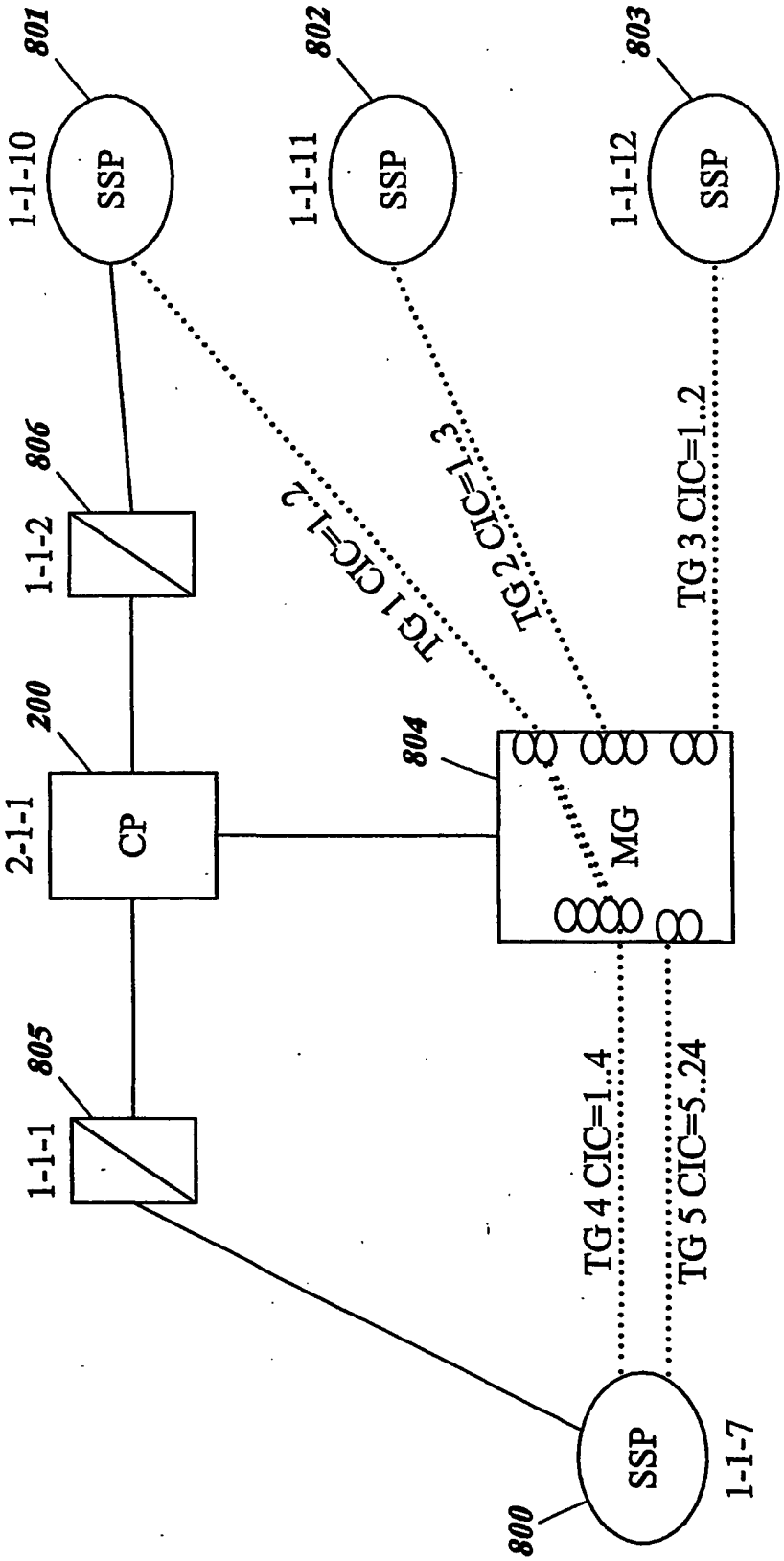
801
802
803
1-1-10
SSP
1-1-11
SSP
1-1-12
SSP
806
1-1-2
TG 1 CIC=1..2
TG 2 CIC=1..3
TG 3 CIC=1..2
200
804
2-1-1
CP
MG
805
1-1-1
TG 4 CIC=1..4
TG 5 CIC=5..24
SSP
1-1-7
800

FIG. 8

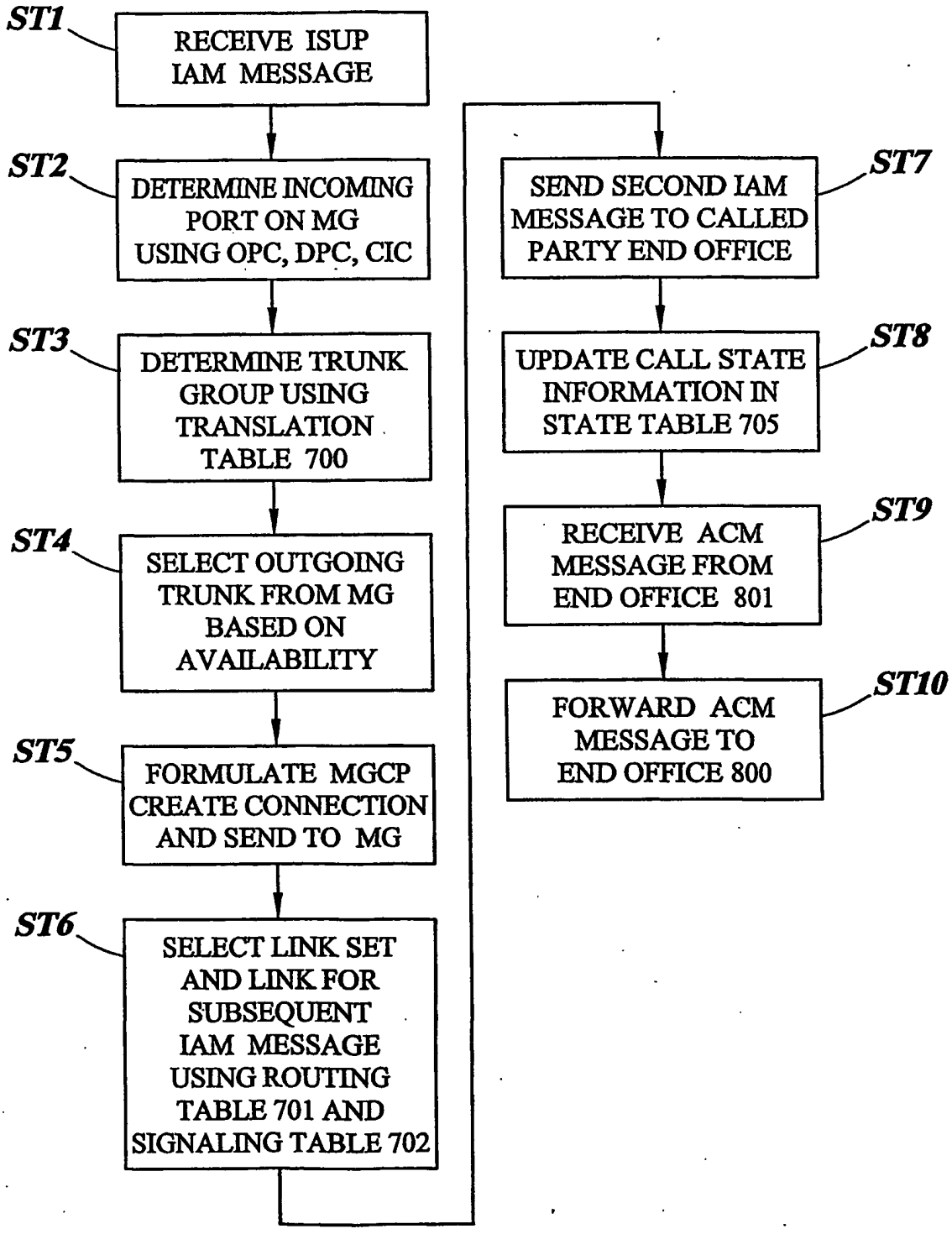
ST1
RECEIVE ISUP IAM MESSAGE
ST2
DETERMINE INCOMING PORT ON MG USING OPC, DPC, CIC
ST3
DETERMINE TRUNK GROUP USING TRANSLATION TABLE 700
ST4
SELECT OUTGOING TRUNK FROM MG BASED ON AVAILABILITY
ST5
FORMULATE MGCP CREATE CONNECTION AND SEND TO MG
ST6
SELECT LINK SET AND LINK FOR SUBSEQUENT IAM MESSAGE USING ROUTING TABLE 701 AND SIGNALING TABLE 702
ST7
SEND SECOND IAM MESSAGE TO CALLED PARTY END OFFICE
ST8
UPDATE CALL STATE INFORMATION IN STATE TABLE 705
ST9
RECEIVE ACM MESSAGE FROM END OFFICE 801
ST10
FORWARD ACM MESSAGE TO END OFFICE 800

FIG. 9

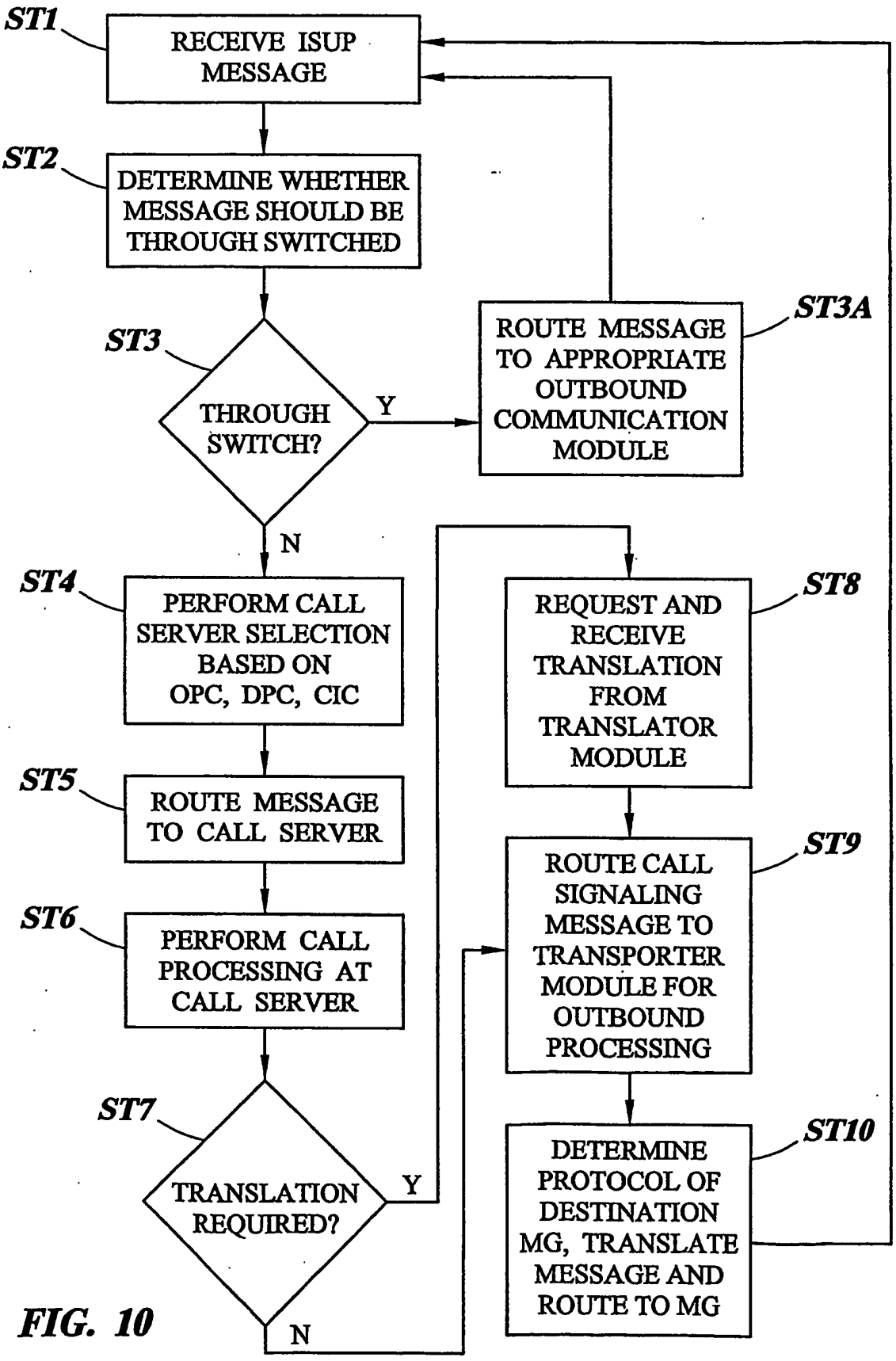
ST1
RECEIVE ISUP MESSAGE
ST2
DETERMINE WHETHER MESSAGE SHOULD BE THROUGH SWITCHED
ST3
THROUGH SWITCH?
Y
ST3A
ROUTE MESSAGE TO APPROPRIATE OUTBOUND COMMUNICATION MODULE
N
ST4
PERFORM CALL SERVER SELECTION BASED ON OPC, DPC, CIC
ST5
ROUTE MESSAGE TO CALL SERVER
ST6
PERFORM CALL PROCESSING AT CALL SERVER
ST7
TRANSLATION REQUIRED?
Y
N
ST8
REQUEST AND RECEIVE TRANSLATION FROM TRANSLATOR MODULE
ST9
ROUTE CALL SIGNALING MESSAGE TO TRANSPORTER MODULE FOR OUTBOUND PROCESSING
ST10
DETERMINE PROTOCOL OF DESTINATION MG, TRANSLATE MESSAGE AND ROUTE TO MG

FIG. 10

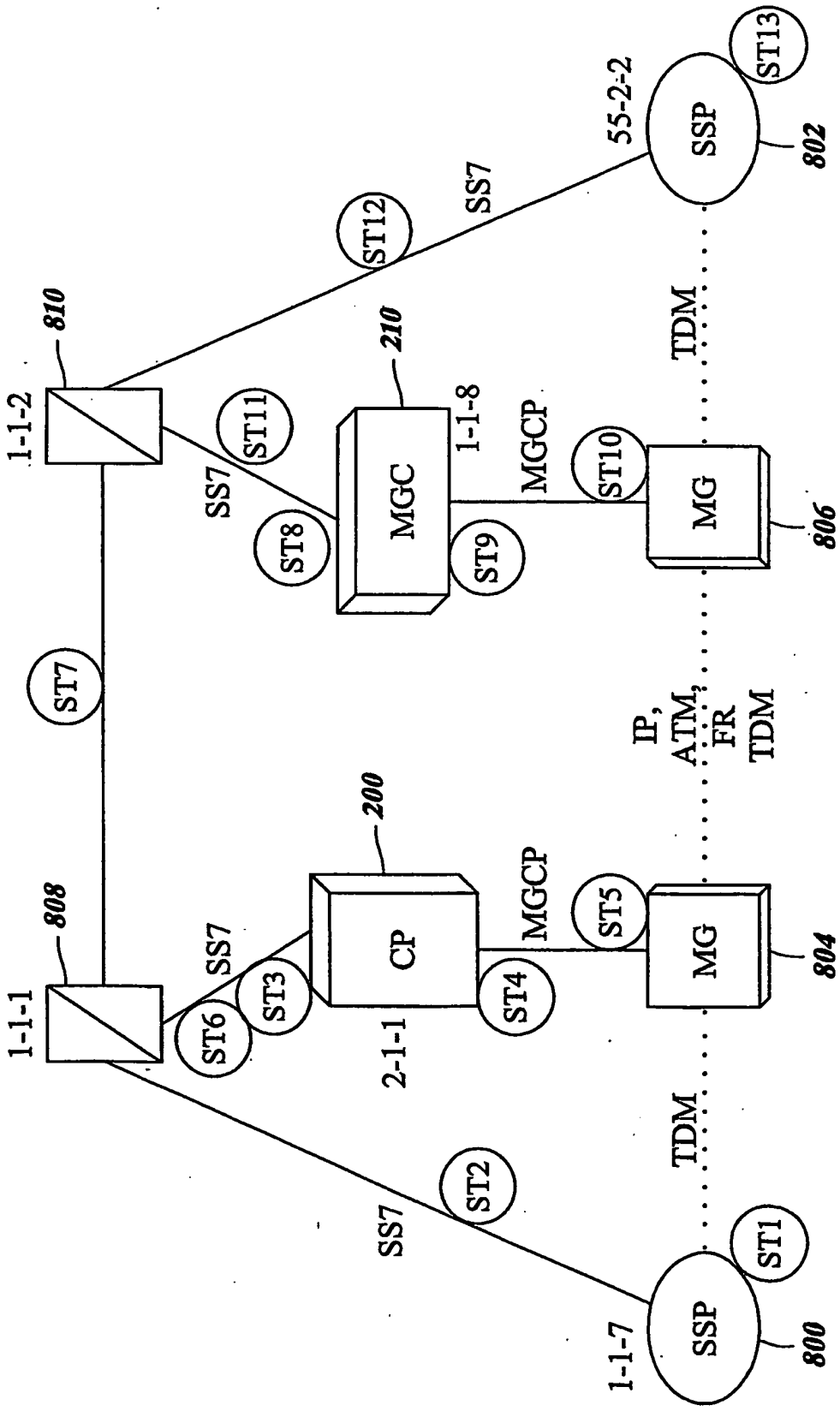
SSP
800
1-1-7
ST1
TDM
ST2
SS7
1-1-1
808
ST6
SS7
ST3
CP
200
2-1-1
ST4
MGCP
ST5
MG
804
IP,
ATM,
FR
TDM
ST7
1-1-2
810
ST11
SS7
ST8
MGC
210
1-1-8
ST9
MGCP
ST10
MG
806
TDM
ST12
SS7
55-2-2
SSP
802
ST13

FIG. 11

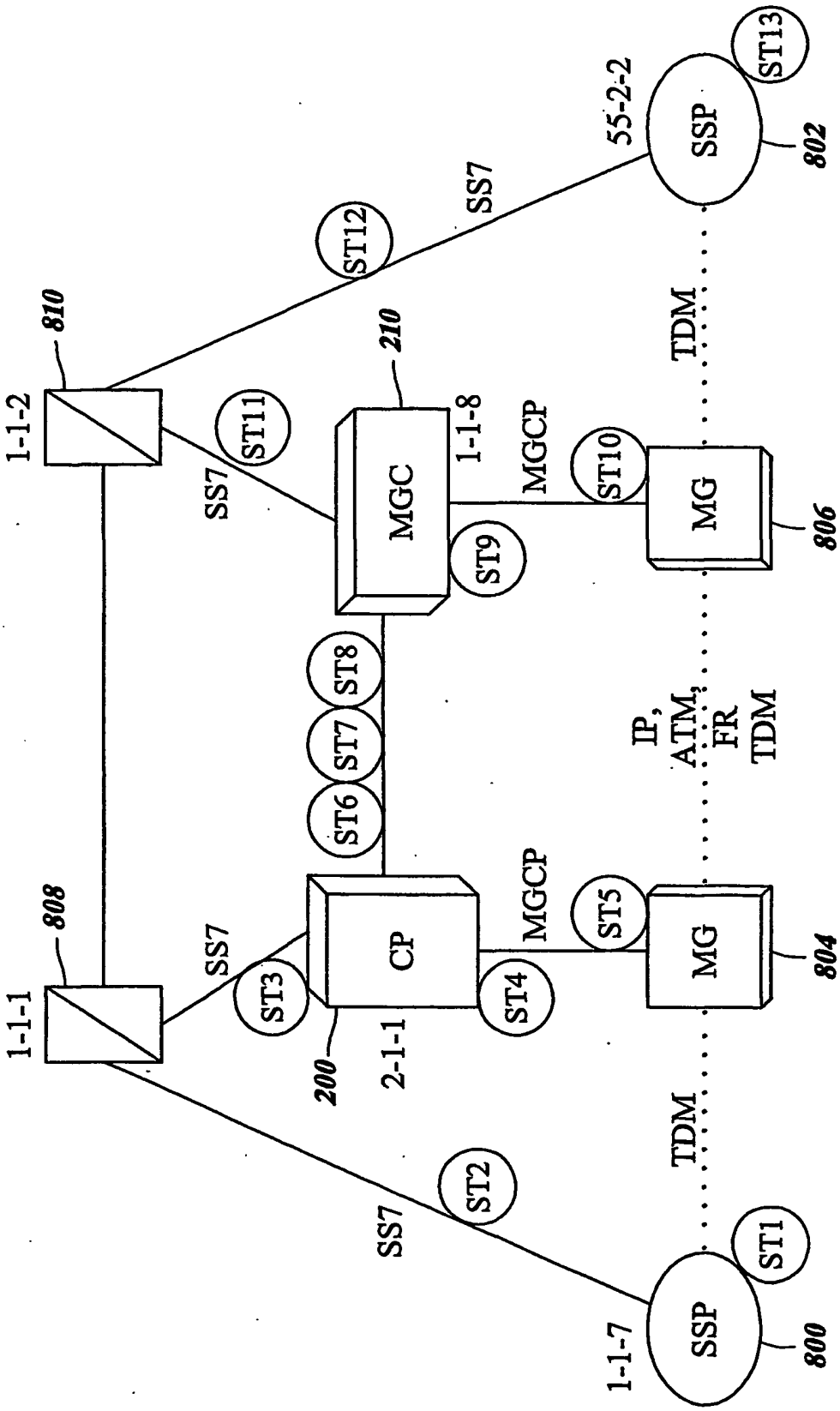
SSP
800
1-1-7
ST1
ST2
SS7
TDM
SSP
802
55-2-2
ST13
SS7
ST12
810
1-1-2
808
1-1-1
SS7
ST11
MGC
210
1-1-8
MGCP
ST9
ST10
MG
806
TDM
IP,
ATM,
FR
TDM
ST8
ST7
ST6
SS7
ST3
CP
200
2-1-1
MGCP
ST4
ST5
MG
804

FIG. 12

INTERNATIONAL SEARCH REPORT

Internation plication No
PCT/US 01/42019

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04Q3/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04Q

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)
EPO-Internal, WPI Data, PAJ, INSPEC

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	POLYZOIS C A ET AL: "FROM POTS TO PANS: A COMMENTARY ON THE EVOLUTION TO INTERNET TELEPHONY" IEEE NETWORK, IEEE INC. NEW YORK, US, vol. 13, no. 3, May 1999 (1999-05), pages 58-64, XP000870632 ISSN: 0890-8044 the whole document	1-33
A	HUITEMA AND AL C: "AN ARCHITECTURE FOR RESIDENTIAL INTERNET TELEPHONY SERVICES" IEEE INTERNET COMPUTING, IEEE SERVICE CENTER, PISCATAWAY, NJ, US, 1 May 1999 (1999-05-01), pages 73-77, XP002144528 ISSN: 1089-7801 the whole document	1-33

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[X] Further documents are listed in the continuation of box C. [ ] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance
"E" earlier document but published on or after the international filing date
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"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed
"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.
"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
5 February 2002	13/02/2002

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Chassatte, R

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INTERNATIONAL SEARCH REPORT

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C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	SIMEONOV P L ET AL: "INGATE: A DISTRIBUTED INTELLIGENT NETWORK APPROACH TO BRIDGE SWITCHING AND PACKET NETWORKS" PROCEEDINGS OF THE INTERNATIONAL CONFERENCE ON COMPUTER COMMUNICATIONS AND NETWORKS, XX, XX, 1997, pages 358-363, XP002073675 the whole document	

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